AMENDMENTS TO THE CLAIMS

Listing of Claims:

1. (Currently amended) A method for testing the transmission quality of a bidirectional real speech transmission or multicast connection over an IP network between a first VoIP endpoint and a second VoIP endpoint, comprising:

transmitting a first number of RTP speech packets in a direction of the second VoIP endpoint, and transmitting a second number of the RTP speech packets in a direction of the first VoIP endpoint; and

detecting at a detection point on a transmission channel between the first and the second VoIP endpoint over a predetermined time period, and arithmetically processing the first and second numbers, and outputting a value representing the transmission quality, wherein the arithmetic processing includes one of a division, where a value 1 of the quotient represents the highest transmission quality and a subtraction, where a value 0 for the difference represents the highest transmission quality.

2. (Previously presented) The method as claimed in claim 1, wherein the predetermined time period for a 10 Mbit/s transmission channel is longer that 5 s.

3. (Canceled)

4. (Canceled)

5. (Previously presented) The method as claimed in claim 1, wherein the value representing the transmission quality is subjected to a threshold value discrimination to suppress side effects due to features of a communication protocol.

6. (Previously presented) The method as claimed in claim 1, wherein quotients outside a predetermined tolerance range around the value 1 are valid as a representation of a reduced transmission quality.

7. (Previously presented) The method as claimed in claim 1, wherein difference values outside a predetermined tolerance range around the value 0 are valid as a representation of a reduced transmission quality.

8. (Previously presented) The method as claimed in claim 1, wherein the detected first and second numbers and/or the calculated values for a plurality of first and second VoIP endpoints connected to the IP network between which bidirectional speech connections exist in each case are logged.

9. (Previously presented) The method as claimed in claim 1, wherein the detected first and second numbers and/or the calculated values for a plurality of first and second VoIP endpoints connected to the IP network within which bidirectional speech connections exist in each case are subjected to summarizing statistical processing to obtain an overall value representing the overall transmission quality of the IP network or of a section of the overall transmission quality of the IP Network.

10. (Previously presented) The method as claimed in claim 1, wherein the value representing the transmission quality is signaled to subscribers at the first and/or second VoIP endpoints and/or to an operation control center of the IP network.

11. (Previously presented) The method as claimed in claim 1, wherein the value representing the transmission quality is used as an input variable for controlling the speech transmission over the IP network.

12. (Previously presented) The method as claimed in claim 1, wherein the value representing the transmission quality is determined substantially in real time and is signaled or is used as an input variable for controlling the speech transmission.

13. (Previously presented) The method according to claim 2, wherein the predetermined time period is in the range of about 10 s to 30 s.

14. (Currently Amended) A method for controlling a speech transmission over an IP network between a first VoIP endpoint and a second VoIP endpoint, comprising:

transmitting a first number of RTP speech packets in the direction of the second VoIP endpoint, and transmitting a second number of the RTP speech packets in the direction of the first VoIP endpoint; and

detecting at a detection point on a transmission channel between the first and the second VoIP endpoint over a predetermined time period, and arithmetically processing the first and second numbers, and outputting a value representing the transmission quality; and

routing the connection between the first and second VoIP endpoints based on the value, wherein the arithmetic processing includes <u>one of</u> a division, where a value 1 of the quotient represents the highest transmission quality and a subtraction, where a value 0 for the difference represents the highest transmission quality.

15. (Currently amended) A method for controlling a speech transmission over an IP network between a first VoIP endpoint and a second VoIP endpoint, comprising:

transmitting a first number of RTP speech packets in the direction of the second VoIP endpoint,

transmitting a second number of the RTP speech packets in the direction of the first VoIP endpoint;

detecting at a detection point on a transmission channel between the first and the second VoIP endpoint over a predetermined time period, and arithmetically processing the first and second numbers, and outputting a value representing the transmission quality; and

setting transmission parameters based on the value, wherein the arithmetic processing includes <u>one of</u> a division, where a value 1 of the quotient represents the highest transmission quality and a subtraction, where a value 0 for the difference represents the highest transmission quality.

16. (Currently amended) A system, comprising:

a detecting unit, arranged at a detection point on a transmission channel between a first and a second VoIP endpoints to detect a first number of RTP speech packets transmitted in a direction of the second VoIP endpoint, and to detect a second number of the RTP speech packets transmitted in a direction of the first VoIP endpoint; and

an arithmetic processing unit connected on the input side to the detecting unit to calculate a value representing the transmission quality from the first and second numbers, wherein the arithmetic processing includes <u>one of</u> a division, where a value 1 of the quotient represents the

highest transmission quality and a subtraction, where a value 0 for the difference represents the highest transmission quality.

17. (Previously presented) The system as claimed in claim 16, wherein the arithmetic processing unit has a division or subtraction stage.

18. (Previously presented) The system as claimed in claim 16, wherein connected downstream of the arithmetic processing unit is a threshold value discriminator to evaluate the value representing the transmission quality with the aid of at least one predetermined threshold value.

19. (Previously presented) The system as claimed in claim 16, further comprising a storage device connected on the input side to the output of the detecting device and/or of the arithmetic processing unit to log the first and second numbers and/or the calculated values.

20. (Previously presented) The system as claimed in claim 16, further comprising a statistical processing unit, connected on the input side to the output of the detecting device and/or of the arithmetic processing unit, to summarize statistical processing of the detected numbers or calculated values in order to evaluate the overall transmission quality of the IP network or of a section of the same.

21. (Previously presented) The system as claimed in claim 16, further comprising a signaling device to signal the calculated value or the overall value to the subscribers at the first and/or second VoIP endpoint and/or to an operation control center of the IP network.